## AMENDMENTS TO THE CLAIMS:

Please amend the claims as follows:

1.(original): A voice code conversion apparatus to which voice code obtained by a first voice encoding method is input for converting this voice code to voice code of a second voice encoding method, comprising:

code separating means for separating, from the voice code based upon the first voice encoding method, codes of a plurality of components necessary to reconstruct a voice signal;

dequantizers for dequantizing the codes of each of the components and outputting dequantized values;

quantizers for quantizing the dequantized values, which are output from respective ones of said dequantizers, by the second voice encoding method to generate codes; and

means for multiplexing the codes output from respective ones of said quantizers and outputting voice code based upon the second voice encoding method.

2.(original): A voice code conversion apparatus, in which a fixed number of samples of a voice signal. is adopted as one frame, for obtaining a first LPC code obtained by quantizing linear prediction coefficients (LPC coefficients), which are obtained by frame-by-frame linear prediction analysis, or LSP parameters found from these LPC coefficients; a first pitch-lag code, which specifies an output signal of an adaptive codebook that is for outputting a periodic sound-source signal; a first noise code, which specifies an output signal of a noise codebook that is for outputting a noisy sound-source signal; and a first gain code obtained by quantizing adaptive codebook gain, which represents amplitude of the output signal. of the adaptive codebook, and noise codebook gain, which represents amplitude of the output signal of the noise codebook; wherein a method for encoding the voice signal by these codes is assumed to be a first voice

encoding method and a method for encoding the voice signal by a second LPC code, a second pitch-lag code, a second noise code and a second gain code, which are obtained by quantization in accordance with a quantization method different from that of the first voice encoding method, is assumed to be a second voice encoding method; and wherein voice code that has been encoded by the first voice encoding method is input to said apparatus for being converted to voice code of the second voice encoding method; said apparatus comprising:

LPC code conversion means for dequantizing the first LPC code by an LPC dequantization method according to the first voice encoding method, and quantizing the dequantized values of LPC coefficients using an LPC quantization table according to the second voice encoding method to find the second LPC code;

pitch-lag conversion means for converting the first pitch-lag code to the second pitch-lag code by conversion processing that takes into consideration a difference between the pitch-lag code according to the first voice encoding method and the pitch-lag code according to the second voice encoding method;

noise code conversion means for converting the first noise code to the second noise code by conversion processing that takes into consideration a difference between the noise code according to the first voice encoding method and the noise code according to the second voice encoding method;

gain dequantization means for dequantizing the first gain code by a gain dequantization method according to the first voice encoding method to thereby find a gain dequantized value; and

gain code conversion means for quantizing the gain dequantized value using a gain quantization table according to the second voice encoding method to convert the gain dequantized value to the second gain code.

3.(original): The apparatus according to claim 2, wherein said gain dequantization means finds a dequantized value of adaptive codebook gain and a dequantized value of noise

codebook gain by dequantizing the first gain code by the gain dequantization method according to the first voice encoding method; and

said gain code conversion means generates adaptive codebook gain code and noise codebook gain code by separately quantizing the dequantized values of the adaptive codebook gain and noise codebook gain using the gain quantization table according to the second voice encoding method, and constructs the second gain code from these two gain codes.

4.(original): The apparatus according to claim 3, wherein said gain code conversion means includes:

first gain code converting means for generating the adaptive codebook gain code by quantizing the dequantized value of adaptive codebook gain using the gain quantization table according to the second voice encoding method; and

second gain code converting means for generating the noise codebook gain code by quantizing the dequantized value of noise codebook gain using the gain quantization table according to the second voice encoding method.

5.(original): The apparatus according to claim 2, wherein frame length according to the first voice encoding method is half the frame length according to the second voice encoding method, a frame according to the first voice encoding method includes two subframes, a frame according to the second voice encoding method includes four subframes, the first voice encoding method expresses pitch-lag code by $n_0$, $n_1$ bits subframe by subframe and the second voice encoding method expresses pitch-lag code by $n_0$, $(n_1+1)$, $n_0$, $(n_1+1)$ bits subframe by subframe, and said pitch-lag code conversion means converts the first pitch-lag code to the second pitch- lag code by:

creating four consecutive subframes, in which pitch-lag code is expressed successively by the $n_0$, $n_1$, $n_0$, $n_1$ bits, from two consecutive frames according to the first voice encoding method;

adopting said pitch-lag codes of the first and third subframes as pitch-lag codes of the first and third subframes according to the second voice encoding method; and

adopting pitch-lag codes, which are obtained by adding a constant value to said pitch-lag codes of the second and fourth subframes, as pitch-lag codes of the second and fourth subframes of the second voice encoding method.

6.(original): The apparatus according to claim 2, wherein frame length according to the first voice encoding method is half the frame length according to the second voice encoding method, a frame according to the first voice encoding method includes two subframes, a frame according to the second voice encoding method includes four subframes, the first voice encoding method expresses noise code by $m_1$, $m_1$ bits subframe by subframe and the second voice encoding method expresses noise code by $m_1$, $m_1$, $m_1$, $m_1$ bits subframe by subframe, and said noise code conversion means converts the first noise code to the second noise code by:

creating four consecutive subframes, in which noise code is expressed successively by the $m_1$, $m_1$, $m_1$, $m_1$ bits, from two consecutive frames according to the first voice encoding method; and

adopting said noise codes of the first to fourth subframes as noise codes of the first to fourth subframes according to the second voice encoding method.

7.(original): The apparatus according to claim 2, wherein said LPC code conversion means includes:

a first arithmetic unit for calculating a first distance between a dequantized value of the first LPC code and a dequantized value of the second LPC code that has been found;

an interpolator for interpolating a dequantized value of an intermediate second LPC code using a dequantized value of the second LPC code of a present frame and a dequantized value of the second LPC code of the previous frame;

a second arithmetic unit for calculating a second distance between a dequantized value of an intermediate first LPC code and a dequantized value of the intermediate second LPC code that has been found by the interpolation; and

an encoder for encoding dequantized values of the 20 LPC coefficients to the second LPC codes so as to minimize the sum of the first and second distances.

8.(original): The apparatus according to claim 7, further comprising weighting means for weighting the first and second distances, wherein said encoder encodes the dequantized values of the LPC coefficients to the second LPC codes so as to minimize the sum of the weighted first and second distances.

9.(original): The apparatus according to claim 8, wherein said LPC code conversion means includes:

code candidate calculation means which, when LPC coefficients are expressed by an n-order vector and the n-order vector is divided into a plurality of small vectors, is for calculating a plurality of code candidates, for which the sum of the first and second distances is small, on a per-small-vector basis; and

LPC code decision means which, when codes are selected one at a time from among the plurality of code candidates of each small vector and are adopted as an n- order LPC code of LPC coefficient deguantized values, is for deciding an n-order LPC code for which the sum of the first and second distances will be minimized and adopting this LPC code as the second LPC code.

10.(original): A voice code conversion apparatus, in which a fixed number of samples of a voice signal is adopted as one frame, for obtaining a first LPC code obtained by quantizing linear prediction coefficients (LPC coefficients), which are obtained by frame-by-frame linear prediction analysis, or LSP parameters found from these LPC coefficients; a first pitch-lag code,

which specifies an output signal of an adaptive codebook that is for outputting a periodic sound-source signal; a first noise code, which specifies an output signal of a noise codebook that is for outputting a noisy soundsource signal; a first adaptive codebook gain code obtained by quantizing adaptive codebook gain, which represents amplitude of the output signal of the adaptive codebook; and a first noise codebook gain code obtained by quantizing noise codebook gain, which represents amplitude of the output signal of the noise codebook; wherein a method for encoding the voice signal by these codes is assumed to be a first voice encoding method and a method for encoding the voice signal by a second LPC code, a second pitchlag code, a second noise code and a second gain code, which are obtained by quantization in accordance with a quantization method different from that of the first voice encoding method, is assumed to be a second voice encoding method; and wherein voice code that has been encoded by the first voice encoding method is input to said apparatus for being converted to voice code of the second voice encoding method; said apparatus comprising:

LPC code conversion means for dequantizing the first LPC code by an LPC dequantization method according to the first voice encoding method, and quantizing the dequantized values of LPC coefficients using an LPC

quantization table according to the second voice encoding method to find the second LPC code;

pitch-lag conversion means for converting the first pitch-lag code to the second pitch-lag code by conversion processing that takes into consideration a difference between the pitch-lag code according to the first voice encoding method and the pitch-lag code according to the second voice encoding method;

noise code conversion means for converting the first noise code to the second noise code by conversion processing that takes into consideration a difference between the noise code according to the first voice encoding method and the noise code according to the second voice encoding method; and

gain code conversion means for generating the second gain code by collectively quantizing, using a gain quantization table according to the second voice encoding method, a dequantized value obtained by dequantizing the first adaptive codebook gain code by a gain dequantization method according to the first voice encoding method, and a dequantized value obtained by dequantizing the first noise codebook gain code by the gain dequantization method according to the first voice encoding method.

11.(original): The apparatus according to claim 10, wherein said LPC code conversion means includes:

a first arithmetic unit for calculating a first distance between a dequantized value of the first LPC code and a dequantized value of the second LPC code that has been found;

an interpolator for interpolating a dequantized value of an intermediate second LPC code using a dequantized value of the second LPC code of a present frame and a dequantized value of the second LPC code of the previous frame;

a second arithmetic unit for calculating a second distance between a dequantized value of an intermediate first LPC code and a dequantized value of the intermediate second LPC code that has been found by the interpolation; and

an encoder for encoding dequantized values of the 5 LPC coefficients to the second LPC codes so as to minimize the sum of the first and second distances.

12.(original): The apparatus according to claim 11, further comprising weighting means for weighting the first and second distances, wherein said encoder encodes the dequantized values of the LPC coefficients to the second LPC code so as to minimize the sum of the weighted first and second distances.

13.(original): The apparatus according to claim 12, wherein said LPC code conversion means includes:

code candidate calculation means which, when LPC coefficients or LSP parameters are expressed by an n-order vector and the n-order vector is divided into a plurality of small vectors, is for calculating a plurality of code candidates, for which the sum of the first and second distances is small, on a per-small-vector basis; and

LPC code decision means which, when codes are selected one at a time from among the plurality of code candidates of each small vector and are adopted as an n-order LPC code of LPC coefficient dequantized values, is for deciding an n-order LPC code for which the sum of the first and second distances will be minimized and adopting this LPC code as the second LPC code.

14.(original): The apparatus according to claim 10, wherein frame length according to the first voice encoding method is twice the frame length according to the second voice encoding method, a frame according to the first voice encoding method includes four subframes, a frame according to the second voice encoding method includes two subframes, the first voice encoding method expresses pitch-lag code by $n_0$, $(n_1+1)$, $n_0$, $(n_1+1)$ bits subframe by subframe and the second voice encoding method expresses pitch-lag code by $n_0$, $n_1$ bits subframe by subframe, and said pitch-lag code conversion means converts the first pitch-lag code to the second pitch-lag code by:

adopting pitch-lag codes of the first and third subframes, from among the pitch-lag codes expressed by the $n_0$, $(n_1+1)$, $n_0$, $(n_1+1)$ bits in four consecutive subframes according to the first voice encoding method, as pitch-lag codes of first subframes of consecutive first and second frames according to the second voice encoding method; and

adopting pitch-lag codes, which are obtained by subtracting a constant value from the pitch-lag codes of the second and fourth subframes, as pitch-lag codes of second subframes of consecutive first and second frames according to the second voice encoding method.

15.(original): The apparatus according to claim 10, wherein frame length according to the first voice encoding method is twice the frame length according to the second voice encoding method, a frame according to the first voice encoding method includes four subframes, a frame according to the second voice encoding method includes two subframes, the first voice encoding method expresses each of the noise codes of the four subframes by $m_1$, $m_1$, $m_1$, $m_1$ and the second voice encoding method expresses each of the noise codes of the two subframes by $m_1$, $m_1$, and said noise code conversion means converts the first noise code to the second noise code by:

adopting the noise codes of the first and second subframes according to the first voice encoding method as noise codes of first and second subframes of the first frame according to the second voice encoding method; and

adopting the noise codes of the third and fourth subframes according to the first voice encoding method as noise codes of first and second subframes of the second frame according to the second voice encoding method.

16.(currently amended): An acoustic code conversion apparatus to which an acoustic code obtained by encoding an acoustic signal by a first acoustic encoding method frame by frame is input for converting this acoustic code to an acoustic code of a second acoustic encoding method and outputting the latter acoustic code, comprising:

code separating means for separating codes of a plurality of components necessary to reconstruct an acoustic signal from the acoustic code that is based upon the first acoustic encoding method;

dequantizers for dequantizing the separated codes and outputting dequantized values;

quantizers for quantizing the dequantized values, which are output from respective ones of said dequantizers, by the second acoustic encoding method to generate codes; and

~~code conversion means for converting the separated codes of the plurality of components to acoustic codes of the second acoustic encoding method;~~

code correction means for inputting the separated codes to said dequantizers ~~code conversion means~~ if a transmission- path error has not occurred, and inputting codes, which are obtained by applying error concealment processing to the separated codes, to said quantizers ~~code conversion means~~ if a transmission-path error has occurred; and

means for multiplexing the codes output from respective ones of said quantizers ~~code conversion means~~ and outputting an acoustic code that is based upon the second acoustic encoding method.

17.(original): An acoustic code conversation apparatus, in which a fixed number of samples of an acoustic signal are adopted as one frame, for obtaining a first LPC code obtained by quantizing linear prediction coefficients (LPC coefficients), which are obtained by frame-by-frame linear prediction analysis, or LSP parameters found from these LPC coefficients; a first pitch-lag code, which specifies an output signal of an adaptive codebook that is for outputting a periodic sound-source signal; a first algebraic code, which specifies an output signal of an algebraic codebook that is for outputting a noisy sound-source signal; and a first gain code obtained by pitch gain, which represents amplitude of the output signal of the adaptive codebook, and algebraic codebook gain, which represents amplitude of the output signal of the algebraic codebook; wherein a method for encoding the acoustic signal by these codes is assumed to be a first acoustic encoding method and a method for encoding the acoustic signal by a second LPC code, a second pitch-lag code, a second algebraic code and a second gain code, which are obtained by quantization in accordance with a quantization method different from that of the first acoustic encoding method, is assumed to be a second acoustic encoding method; and wherein acoustic code that has been encoded by the first acoustic encoding method is input to said apparatus for being converted to acoustic code of the second acoustic encoding method; said apparatus comprising:

code separating means for separating codes of a plurality of components necessary to reconstruct an acoustic signal from the acoustic code that is based upon the first acoustic encoding method;

code conversion means for converting the separated codes of the plurality of components to acoustic codes of the second acoustic encoding method;

code correction means for inputting the separated codes to said code conversion means if a transmission- path error has not occurred, and inputting codes, which are obtained by applying error concealment processing to the separated codes, to said code conversion means if a transmission-path error has occurred; and

means for multiplexing the codes output from respective ones of said code conversion means and outputting an acoustic code that is based upon the second acoustic encoding method.

18.(original): The apparatus according to claim 17, wherein if a transmission-path error has occurred in the present frame, said error correction means estimates an LPC dequantized value of the present frame by an LPC dequantized value of a past frame, and said code conversion means finds, from the estimated LPC dequantized value, the LPC code in the present frame that is based upon the second acoustic encoding method.

19.(original): The apparatus according to claim 17, wherein if a transmission-path error has occurred in the present frame, said error correction means executes the error concealment processing by adopting a past pitch-lag code as the pitch-lag code of the present frame, and said code conversion means finds, from the past pitch-lag code, the pitch-lag code in the present frame that is based upon the second acoustic encoding method.

20.(original): The apparatus according to claim 17, wherein if a transmission-path error has occurred in the present frame, said error correction means executes the error concealment

processing by adopting a past algebraic code as the algebraic code of the present frame, and said code conversion means finds, from the past algebraic code, the algebraic code in the present frame that is based upon the second acoustic encoding method.

21.(original): The apparatus according to claim 17, wherein if a transmission-path error has occurred in the present frame, said error correction means estimates a gain code of the present frame by a past gain code, and said code conversion means finds, from the estimated gain code, the gain code in the present frame that is based upon the second acoustic encoding method.

22.(original): The apparatus according to claim 17, wherein if a transmission-path error has occurred in the present frame, said error correction means finds a pitch gain Ga obtained from a dequantized value of past pitch-gain and finds an algebraic codebook gain Gc obtained from a dequantized value of past algebraic codebook gain, and said code conversion means finds, from this pitch gain Ga and algebraic codebook gain Gc, the gain code in the present frame that is based upon the second acoustic encoding method.